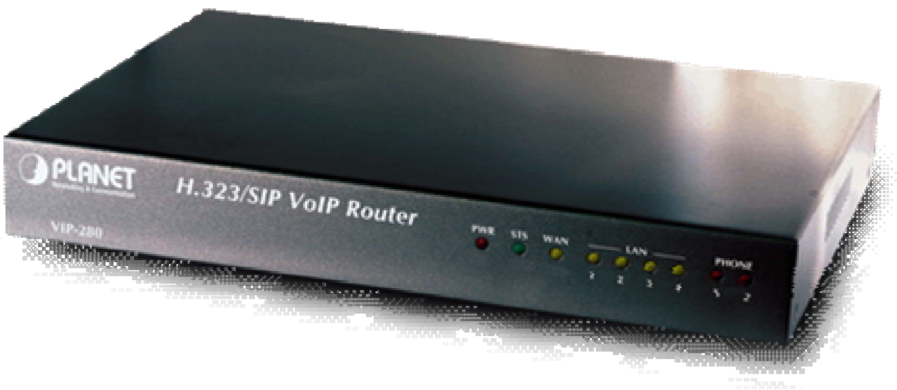
H.323/SIP VoIP Router

VIP-280

User's manual



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CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

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Revision

User’s Manual for PLANET H.323/SIP VoIP Router:

Model: VIP-280

Rev: 1.0 (Sept. 2005)

Part No. EM-VIP280V1

Chapter 1 Introduction

1

Overview

With years of Internet telephony and router manufacturing experience, PLANET proudly introduces the latest member of the PLANET VoIP gateway family: the VIP-280.

According to the feedbadks from our customers, PLANET's new VoIP gateway, the VIP-280, not only provides high quality voice communications, but also offers secured, reliable Internet sharing capabilities for either daily voice or Internet communications. With advanced DSP processor and cutting edge VoIP technology, the PLANET VIP-280 is capable of handling both SIP and the H.323 calls. Up to 4 registrations to the SIP proxy or H.323 Gatekeeper, the VIP-280 is able to make calls to either H.323 or SIP voice communication environment. Moreover, the VIP-280 is the ideal choice for Voice over IP communications and providing integrated Internet sharing features, such as Virtual server, SPI firewall protection, and DMZ support; with these features, users may now enjoy high quality voice calls and secure Internet access without interfering with routine activities.

With built-in PPPoE / DHCP / DDNS clients, up to 2 concurrent voice connections on machine can be established wherever around the world. The PLANET VIP-280 presents with an intuitive, user-friendly, yet powerful Web management interface, which can dramatically reduce IT personnel resource requirements, and complete VoIP deployment in a short time. With a remote management capability, device administrators can monitor machine/network status, or perform maintenance and trouble-shooting service via an Internet browser or telnet session.

Firewall/Security Feature

- Built in NAT firewall, DoS (Denial of Service) protection
- SPI (Stateful Packet Inspection) firewall
- Policy-based LAN/WAN access control
- Virtual server, DMZ
- Remote administrator authentication
- Enable/disable VPN pass-through

VoIP Functions

- H.323 / SIP dual mode communication
- SIP 2.0 (RFC3261), H.323v3 compliant
- Peer-to-Peer / H.323 GK / SIP proxy calls
- Voice codec support: G.711, G.723.1A, G.729A

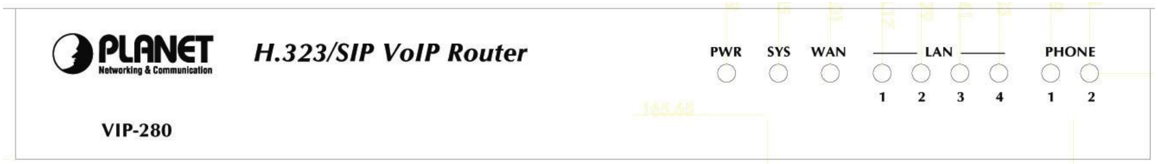
- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation, Call progress detection, Gain Control

Package Content

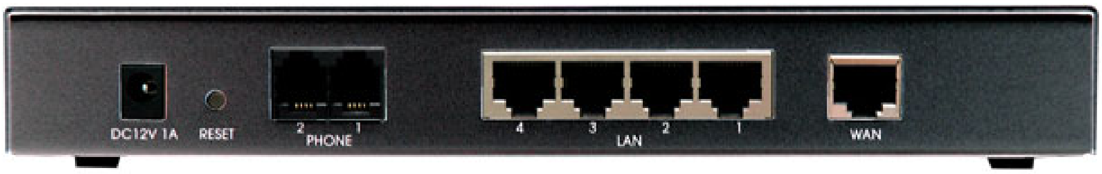
The contents of your product should contain the following items:
H.323/SIP VoIP router
Power adapter
Quick Installation Guide
User's Manual CD
RJ-11 cable x 2

Physical Details

The following figure illustrates the front/rear panel of VIP-280.



Front Panel of VIP-280



Rear Panel of VIP-280

LED Display & Button

LED Indicators	Descriptions
PWR	Power is supplied to the VoIP router
SYS	System LED will be ON when the registration toward the GK/SIP proxy is successful
WAN	Orange: the VoIP router is connected to WAN at 10Mb/s. Green: the VoIP router is connected to WAN at 100Mb/s.
LAN 1 ~ LAN 4	Orange: the VoIP router is connected to LAN at 10Mb/s. Green: the VoIP router is connected to LAN at 100Mb/s.
Phone 1 ~ Phone 2	Off: the line is idle. On: the line is being used.

Back Panels	Descriptions
DC12V	Power Adapter connecter
Reset	Reset to the default setting
Phone 1~ Phone 2	Connect to the RJ-11 phone line
LAN 1 ~ LAN 4	10/100Mbps Ethernet port, used to connect PC or NB
WAN	10/100Mbps Ethernet port, used to connect ADSL or cable modem

Note

Press RESET button on rear panel over 8 seconds or until SYS LED begins to flash, the VoIP Router to this default LAN/WAN IP address and Username/Passowrd function.
When factory reset is completed, the Default LAN IP is http://192.168.0.1 from factory.

Chapter 2

Preparations & Installation

2

Physical Installation Requirement

This chapter illustrates basic installation of VIP-280

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem (for WAN port usage)

Administration Interface

PLANET VIP-280 provides GUI (Web based, Graphical User Interface) for machine management and administration.

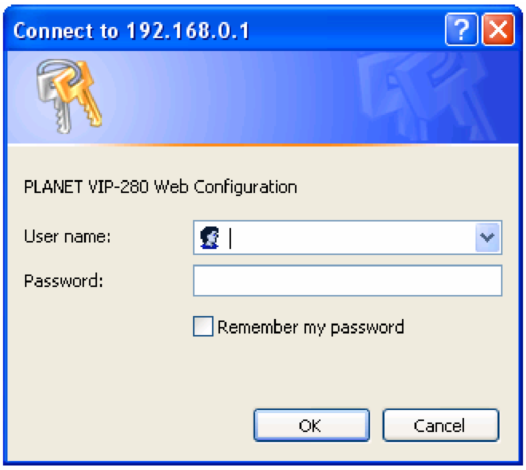
Web configuration access

To start VIP-280 web configuration, you must have one of these web browsers installed on computer for management

- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Default LAN interface IP address of VIP-280 is **192.168.0.1**. You may now open your web browser, and insert **192.168.0.1** in the address bar of your web browser to logon VIP-280 web configuration page.

VIP-280 will prompt for logon username/password, please enter: **admin** / **123** to continue machine administration.



Note

Please locate your PC in the same network segment (192.168.0.x) of VIP-280. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

LAN/WAN Interface quick configurations

Nature of PLANET VIP-280 is an IP Sharing (NAT) device, it comes with two default IP addresses, and default LAN side IP address is "**192.168.0.1**", default WAN side IP address is "**172.16.0.1**". You may use any PC to connect to the LAN port of VIP-280 to start machine administration.

ⓘ Hint

In general cases, the LAN IP address is the default gateway of LAN side workstations for Internet access, and the WAN IP of VIP-280 is the IP address for remote calling party to connect with.

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: **192.168.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **admin** / **123**), browse to "**Administrator**" --> "**LAN setting**" configuration menu:

LAN IP Address :	192	168	0	1
Subnet Mask :	255	255	255	0
Default Gateway :	192	168	0	1

Parameter Description

IP address LAN IP address of VIP-280

Default: 192.168.0.1

Subnet Mask LAN mask of VIP-280

Default: 255.255.255.0

Default Gateway Gateway of VIP-280

Default: 192.168.0.1

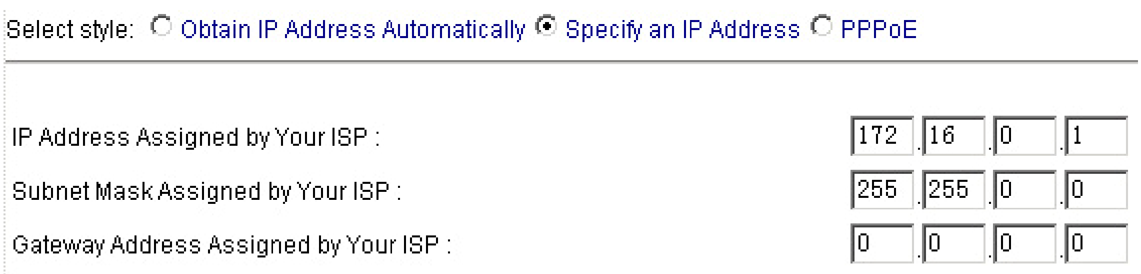
Hint

```
It is suggested to keep the DHCP server related parameters
in default state to keep machine in best performance.
```

After confirming the modification you've done, Please click on the **Modify** button to make the changes effective.

WAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: **172.16.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **admin** / **123**), browse to "**WAN Setting**" configuration menu, you will see the configuration screen below:



Connection Type	Data required.
Obtain IP Address Automatically	In most circumstances, it is no need to configure the DHCP settings.
Specify an IP Address	The ISP will assign IP Address, and related information.
PPPoE	The ISP will assign PPPoE username / password for Internet access,

Hint

```
Please consult your ISP personnel to obtain proper PPPoE/IP
address related information, and input carefully.
If Internet connection cannot be established, please check
the physical connection or contact the ISP service staff
for support information.
```

Save Modification to Flash Memory

Most of the VoIP router parameters will take effective after you modify, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the VoIP router, to save the parameters into Flash ROM and let it take effective forever, please remember to press the **Save Modification** button after you modify the parameters.

Chapter 3

Network Service Configurations

Configuring and monitoring your VIP-280 from web browser

The VIP-280 integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Overview on the web interface of VIP-280

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

To start VIP-280 web configuration, you must have one of these web browsers installed on computer for management

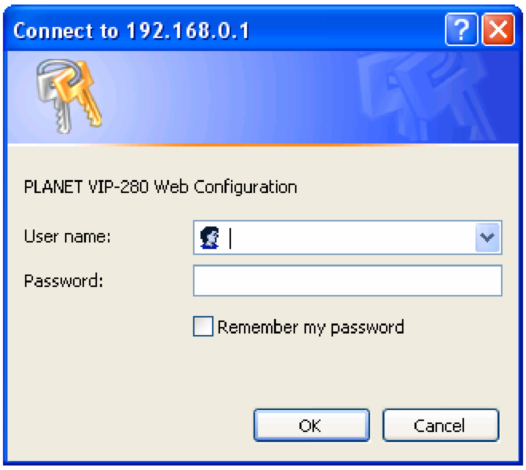
- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Manipulation of VIP-280 via web browser

Log on VIP-280 via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input ***http://192.168.0.1*** to logon VIP-280 web configuration page.

VIP-280 will prompt for logon username/password: ***admin*** / ***123***



VIP-280 log in page

VIP-280 main page

Chapter 4

VoIP Configurations

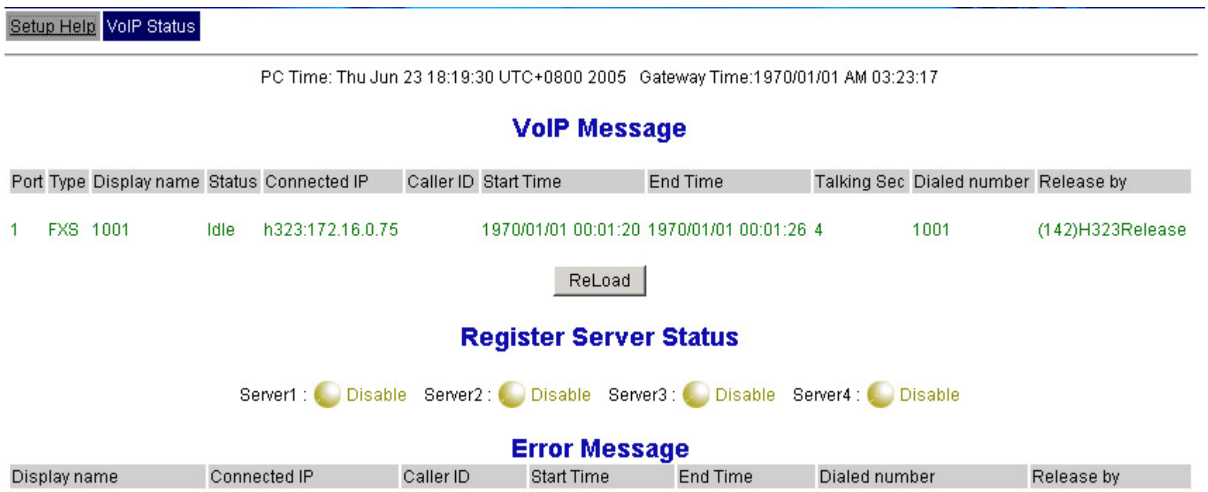
4

VIP-280 Status

This page main display the current and last time VoIP call status & result.

Parameter Description	
PC Time	will show the date & time that your connected PC now.
Gateway Time	will show the date & time of this VoIP router, the date amd time is get from SNTP server. You may setting the SNTP server from **"System Config → Administrator → Date & Time"**
Ports Message	
Port	display FXS interfase the port number.
Type	Telephone interface type: **FXS:** for connect to regulate phone set.
Display Name	display the remote party name of this VoIP call.
Status	Current status of this port.
Idle	Standby make phone call.
Signal	Waiting for DTMF key in or VoIP protocol connecting.
In	There is a phone call made from phone port and call out to Network by VoIP.
Out	There is a phone call made from network VoIP and pick up by phone set.
Connected IP	The other party IP of this VoIP call.
Caller ID	Caller ID received from phone port.
Start Time	Date & time of this VoIP call begin on this port.
End Time	Date & Time of last VoIP call End on this port.
Talking Sec	Total talked seconds of last VoIP call on this port.
Dialed number	On the VoIP call out (line status display In), This will display the real dial out number for VoIP call. On the VoIp call in (line status display out). This will display the number will dial out to phone line.
Release by	This will display the reason of this call termination.

Register Sever Status	This VoIP router can register to 4 GK/SIP proxy simultaneously. Users can setup the GK/SIP proxy information on “**VoIP Config → Register Server**”
Error Message	For some reason (ex. All lines of this VoIP router are busy), here will display the failure information of last time VoIP Call in.

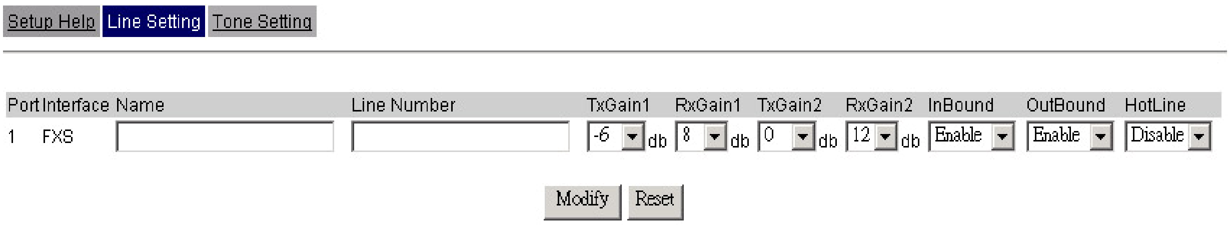


Line Setting

This page will setup the phone line information each port.

Parameter Description	
Port	display FXS interfase the port number.
Interface	Telephone interface type: FXS: for connect to regulate phone set.
Name	Line name for this port. This will send and display on the remote side due VoIP call
Line Number	Telephone number assigned to this line.
TxGain	Transmitter Gain. This will adjust the speaker volume of local phone set. The adjust range is from +3 to -13dB. Higher value will cause louder sound come from local phone set.
RxGain	Receiver Gain. This will adjust the microphone volume of local phone set. The adjust range is from -3 to +13dB. Higher value will cause amplifier the sound get from local phone set.
Inbound	Enable or disable the VoIP call to Internet. Disable the inbound will not allow any call made call to Internet from phone set.
Outbound	Enable or disable the VoIP call from Internet. Disable the Outbound will not allow any call made call from Internet to phone set.

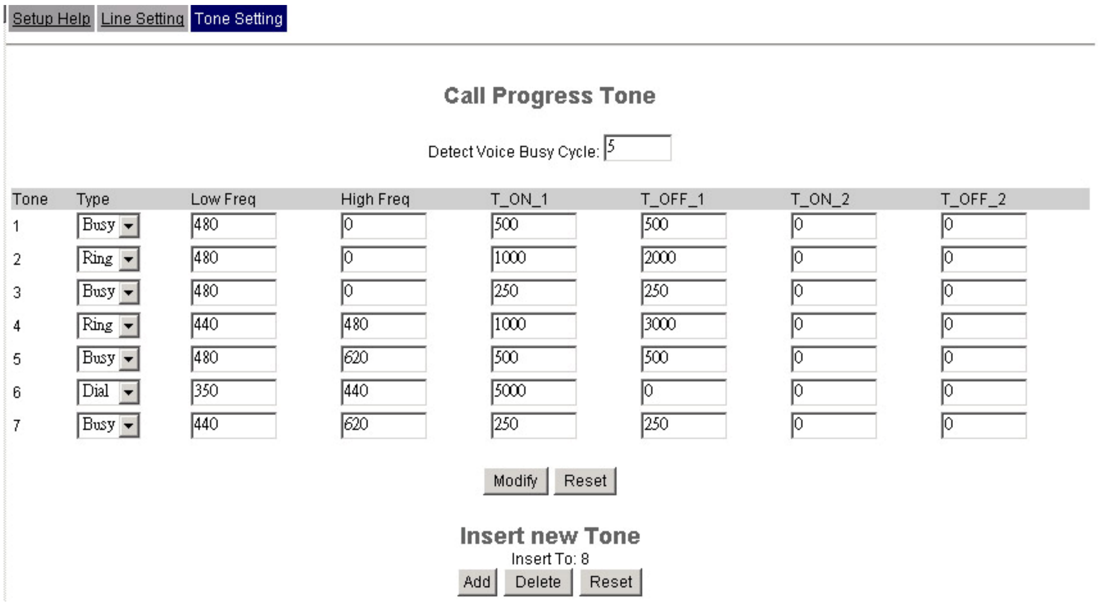
Hotline	When Enable, it will allow you to make a VoIP call without Key in any number. That mean it will direct call out by VoIP when you off hook the phone of this line.



Tone Setting

This page defines the tones generated to the phone connected to the phone port. All lines use same tone parameters. After modify the tone parameters, you must save modify then Reboot to let the modified parameters work.

Parameter Description	
Detect Voice Busy Cycle	Use the parameters to automatic detect cadence busy tone. When detected a voice cadence repeat over this parameters setting in sequence, the VoIP router will treat it like busy tone and disconnect automatically. Please do not set this parameter less than 5 to avoid unexpected erroneous disconnect.
Tone define Table	You can set up to 15 tones set for detection and generation. For the generation, the first entry will be used. The call progress tones, ranging from 300 Hz to 2000 Hz, are defined for both generation and detection. Generation, however, can be defined from 1 Hz to 3980 Hz.
Tone	Maximum 15 tones can be defined.
Type	Dial: Define the generated dial tone for phone set Busy: Define the busy tone for generate & detect Ring: Define the ring back tone for generate
Low freq	Lower frequency for defined tone
High freq	Higher frequency for defined tone. Each tone can define two frequencies, if only one frequency needed, please leave High Frequency to 0.
T_ON_1,T_OFF_1, T_ON_2, T_OFF_2	The cadence pattern of up to four intervals for each dual-frequency. Minimum Cadence value is 30msec.



VoIP Call Out

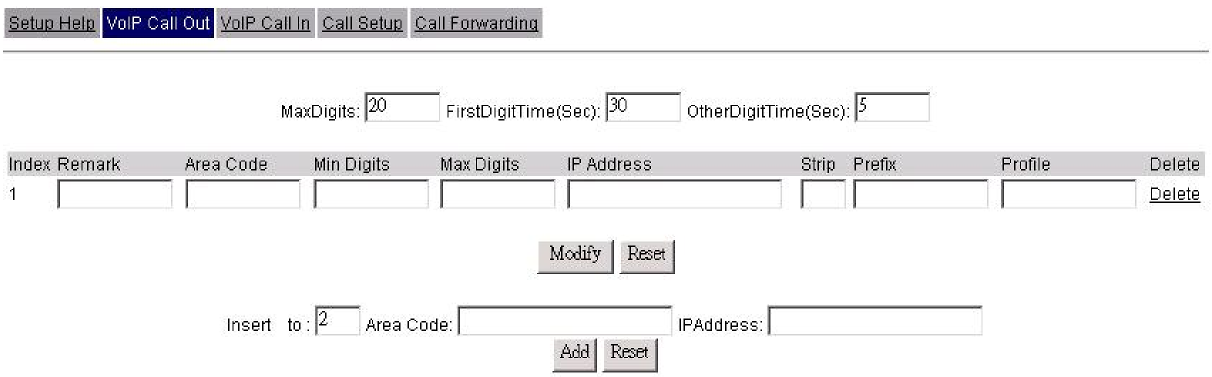
This page defines the routing rule for Call out to VoIP. (User key in the phone number through phone set dial pad, then VoIP router translate the phone number by the routing table setting here to destination IP, and dial out number then call out via network protocol).

Each time when you off hook the phone connected to this VoIP router, you will hear a dial tone to remind you to key in the phone number, after you input the number you called, if digits of the number of you called is not exceed the Max Digits, please remember to press the # key for ending the input.

Parameter Description	
MaxDigits	Define the maximum digits wait for user key in for all VoIP Call Out, if user key in digits match the number defined here. It will go to translate for call out rule without needed to press # key.
FirstDigitTime	Define the waiting seconds for user key in phone number first digit. User need to key in first digits before the seconds defined here, if VoIP router wait over the defined seconds and there is no any digits key in, the VoIP router will feedback the user busy tone.
OtherDigitTime	Define the waiting seconds for user key in phone number secondary & the rest digits. User need to key in the rest digits before the seconds defined here, if VoIP router wait over the defined seconds and there is no any digits key in, the VoIP router will feedback the user busy tone.
Remark	Remark for this routing rule. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.
Area Code	Define the Prefix number fit this rule, any phone number prefix digits

	matched with the rule will call out by this rule define. Please Notify there is a compare order rule on this routing table. That mean the VoIP router will check the rule list from top to bottom one by one, any rule item matched with the prefix digits that user key in will go to call out directly no regard to the rest rules below. For Example, if a rule item for area code 8862 is on Index 5, another rule item for area code 886 on Index 6 below that will be ignored. By setting the **hln** (**hl1** for hot line one, **hl2** for hot line two) on the area code field, and enable hot line function (Please refer to the "**VoIP Config → Line Configure → Line Setting**"), the VoIP router can service the hot line direct call.
Min Digits	Define the minimum digits wait for user key in for number fit this rule, if user key in digits less the number defined here. It will keep waiting for input until exceed the "**FirstDigitTime**" defined time. If user key in digits more then "**Min Digits"** here, the VoIP router will wait time defined on "**OtherDigitTime**" then go to translate for call out rule without needed to press "**#**" key.
Max Digits	Define the maximum digits wait for user key in for number fit this rule, if user key in digits match the number defined here. It will go to translate for call out rule without needed to press "#" key.
IP Address	Define the destination IP for call out number fit this rule, user can input below format: **IP address/URL for H.323 protocol:** such as: **h323:172.16.0.100 / h323:h323.testcall.com.tw** **IP address/URL for SIP protocol:** such as: **sip:172.16.0.200 / sip:sip.testcall.com.tw** **Note:** This H.323/SIP DECT VoIP router can setup to Uregister to DDNS service. (Please refer to the "System Config → Advanced → Dynamic DNS") to let user call out to another VoIP router with dynamic IP by URL. **GK/SIP proxy**, such as: it will get the destination IP by register server setting (Please refer to the "**VoIP Config → Register Server**") in advance.
Strip	The number of digits will be ignored by user input. For example, if user key in the number is 886222199518 and the STRIPE field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 22199518.
Prefix	The numbers will be added on the prefix of user key in number. For examples, if user key in the number is 22199518 and the PREFIX field is setting to 0028862, the actually call out number will be

	002886222199518. Another example, if user key in the number is 90, STRIP field is setting to 2, and the PREFIX field is setting to 0,22199518, the actually call out number will be 0,22199518 (“**,** “ **mean wait 1 second**). This example is especially for speed dial function.
Profile	Define the optional special call out parameters on this destination. Please input the name you Udefined on the profile (Please refer to the “**VoIP Config → Routing Setup → Routing Profile**”) list.
Delete	Delete this rule item on routing table.



To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE and IP address then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the modify button to take it effect. For store back to flash memory, please press ”**Syetem Maintenance → Save Modification**” .

ⓘ Hint

When user enable the hot line function on **“VoIP Config → Line Configure → Line Setting”** menu, it will over ride the above parameters and direct call out by hot line call out rule.

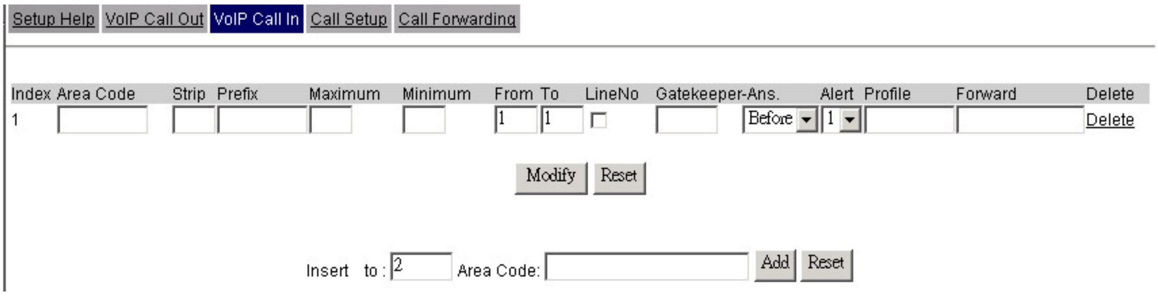
VoIP Call In

This page let you define the routing rule for Call in from VoIP. (VoIP router got a VoIP call required form network, and then translates the phone number passed from remote side VoIP router to the real dial out number, and line base on this VoIP call in routing table). Each time when the VoIP router received a VoIP call from network, it will check with “**Area Code”** to see which rule matched to service, if no rule matched, it will refuse to call out and will bound back the call.

When the VoIP router received a VoIP called from network, it will check below rules fields then decide line and number to dial out.

Parameter Description	
Area Code	Define the Prefix number this rule service, any VoIP called from network dialed number prefix digits matched with the rule will call out to phone by this rule define. Please Notify there is a compare order rule on this routing table. That mean the VoIP router will check the rule list from top to bottom one by one, any rule item matched with the prefix digits that user key in will go to call out directly no regard to the rest rules below. For Example, if a rule item for area code 8862 is on Index 1, another rule below that like index 2 for area code 886 will be ignored.
Strip	Number of digits will be ignored by user input. For example, if received VoIP call number is 886222199518 and the "**STRIPE**" field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 22199518.
Prefix	The numbers will be added on the prefix of received VoIP call number. For examples, if received VoIP call number is 22199518 and the "**PREFIX"** field is setting to 0028862, the actually call out number will be 002886222199518.
Maximum	Define the maximum digits of call number allow to dial. If the length of dial number after pervious "STRIP" and "PREFIX" process is more than the setting, it will deny dialing out. For example, you can set the "**Maximum**" dial out digits is 8, for call to local area phone only, any VoIP call in attempt to dial 0222199518 out of 8 digits for call out long distance will been deny to call out.
Minimum	Define the minimum digits of call number allow to dial. If the length of dial number after pervious "STRIP" and "PREFIX" process is less than the setting, it will deny dialing out. For example, if set "**Minimum**" to 4, any VoIP call in attempt to dial number less than 4 digits like 110, 911 will been deny to call out.
From	Define the beginning line number for service this area code VoIP call. For example, if user assigned FROM 1 TO 1 for AREA CODE 601 in this routing table, then any VoIP call for call in number 601 will ring the line 1 only.
To	Define the ending line number for service this area code VoIP call.
Line No	Click to enable if you want to force compare with the line number setting on ULINE CONFIGUREU menu (Please refer to the "**VoIP Config → Line Config → Line Setting**"). If the dial number after pervious STRIP and PREFIX process is matched with the line number setting, the VoIP

	call will ring the dedicate phone line that assigned with matched number.
Server	Assign which server to authorize this incoming VoIP call before call out. For example, if the dial number should be checked by server 1 setting on the "**Regster Server**" menu (Please refer to the "**VoIP Config → Register Sever**").
ANS	When the call is coming , Before or After to pick up the phone , the Server should check that has the speaker got authorization from Register Server ? **After**：setting on After function , when the call is coming, Server will ring at first, when user pick up the phone, then Server will go to Register Server for checking caller-authorization, if the authorization has confirmed, then the connection will start to success, otherwise it will sent busy tone. **Before**：setting on Before function , when the call is coming, at fist Server will go to Register Server to check that has the speaker got authorization? If the authorization has confirmed, then Server start to ring, otherwise it will send busy tine.
Alert	Control the Ring Back tone generate timing: **Mode 0:** When this VoIP ruter get ring back tone from phone line, it will send the ring Alert signal to remote VoIP router for generate ring back tone. **Mode 1:** Before this VoIP router dial to phone line, it will send the ring Alert signal to remote VoIP router for generate ring back tone. **Mode 2:** After this VoIP router finish dial out number to phone line, it will send Connect OK signal to remote VoIP router. **Mode 3:** Before this VoIP router dial to phone line, it will send the ring Alert signal to remote VoIP router for generate ring back tone, after this VoIP router finish dial out number to phone line, it will send Connect OK signal to remote VoIP router.
Profile	Define the optional special VoIP parameters when received on this destination. Please input the name you defined on the profile list (Please refer to the "**VoIP Config → Call Routoing → Call Setup**").
Forward	Define the profile name for forward the unanswerable VoIP call on this call in rule. Please input the name you defined on the "**Forward**" profile list.
Delete	Delete this rule item on routing table. To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE then press ADD button to add it on the list. Then modify the necessary information on the routing table list.



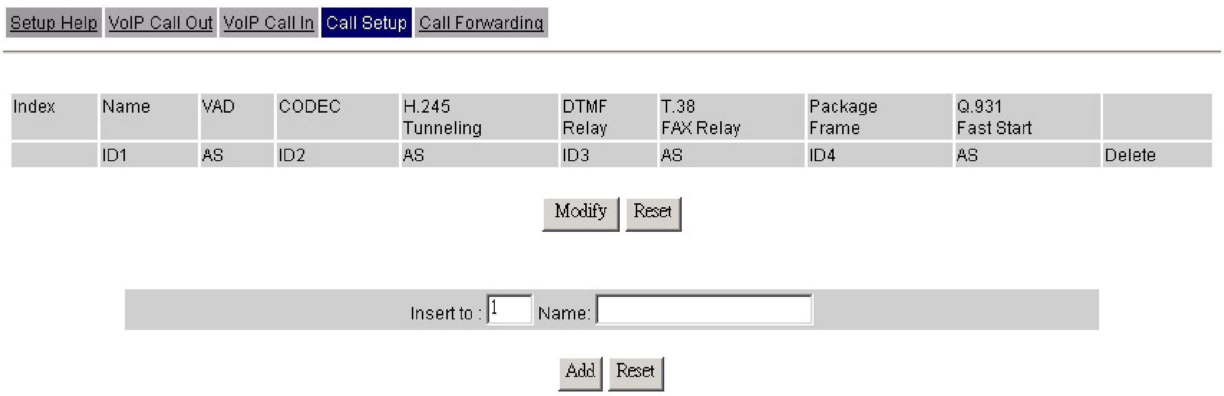
Please remember to press the modify button to take it effect. For store back to flash memory, please press “**Save Modification”** (Plaase refer to the “**Syetem Maintenance → Save Modification**”).

Call Setup

This page defines the optional special VoIP parameters when making/received a VoIP call. For define some special parameters for different VoIP equipment or authorize purpose, please add a profile at “**VoIP Config → Call Routing → Call Setup**”, and use the same name as the profile on the “**Call in Routing Table**” (Please refer to the “**VoIP Config → Call Routing → VoIP Call In**”) or “**Call out Routing table**” (Please refer to the “**VoIP Config → Call Routing → VoIP Call Out**”).

Parameter Description	
Name	Specify a profile name. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.
VAD	**ON:** turn on the VAD (Voice Activity Detection) function. **OFF:** turn off the VAD function, please select ON for save the bandwidth.
CODEC	Select different voice CODEC for VoIP communication. The bit rate of G.723.1 is 5.3k/6.3k, G.729 is 8k, uLaw and aLaw is 64k per second. The G.723.1 is default CODEC.
H.245 tunneling	**ON:** to enable H.245 tunneling. **OFF:** to disable H.245 tunneling.
DTMF Relay	When select UIn bandU to transfer the DTMF during VoIP, the user pressed DTMF tone will be treat as general voice and been compressed then transmit to remote side to decompress play back, it maybe cause some problem on duplicate or missing DTMF receive. When select “**Out band**” to transfer the DTMF during VoIP, the user pressed DTMF tone will be decode by local VoIP router then transmit as signal, after received on received remote VoIP router, it will be regenerate by remote VoIP router. The default value is Out band.

T.38 FAX Relay	**ON:** FAX will be transmitted by using T.38 FAX over IP protocol. **OFF:** FAX over IP is disable.
Package Frame	Select the voice payload frame on each UDP package VoIP transmit. More frames into one package is save more bandwidth. The default frames on each package is 3.
Q.931 Fast Start	**ON:** Enable Fast Start capability during Q.931 handshaking. **OFF:** Disable Fast Start capability during Q.931 handshaking.
ID1	User defines ID #1 during this VoIP call.
As	**E.164:** Parameter on ID1 field is the E.164 during this VoIP call. **H.323 ID:** Parameter on ID1 field is the H.323 ID during this VoIP call. **Calling:** Parameter on ID1 field is DID number during this VoIP call. If this optional is setting, it will override the LINE NUMBER on line Setting menu. **Password:** Parameter on ID1 field is the password for VoIP call. Parameter defined here will used as MD5 during H.235 and will not display on the Web UI
ID2,ID3,ID4	There are 4 fields for user define the ID parameters, please reference the ID1 setting above.
Delete	Delete this rule item on routing table.



To add new profile item on routing table, please assign the number you want to insert before, input profile NAME then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the modify button to take it effect. For store back to flash memory, please press “**Save Modification”** (Plaase refer to the “**Syetem Maintenance → Save Modification**”).

Call Forwarding

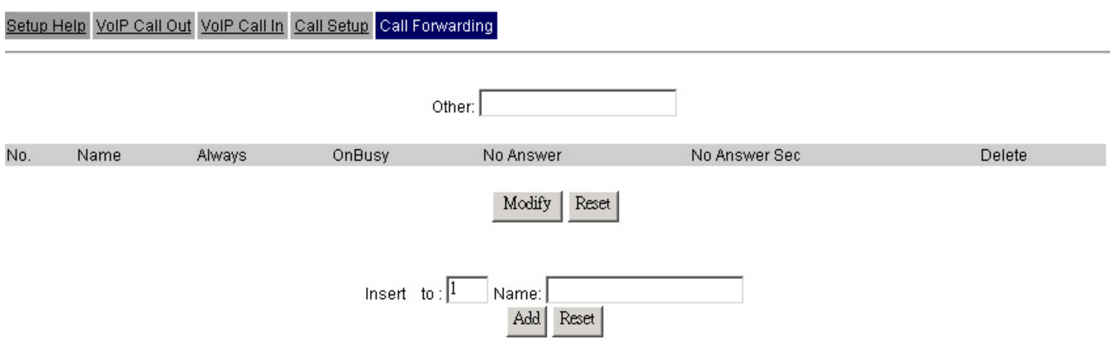
This page defines the scenario of call forwarding:

◆ Get an unmatched prefix number for VoIP call in

- Line busy
- No answer

Please add a profile at “**VoIP Config → Call Routing → Call Setup**” and put the name of profile on the Call out Routing table (Please refer to the “**VoIP Config → Call Routing → /VoIP Call Out**”).

Parameter Description	
Other	Define the forward IP and forward phone number when there is no match rule setting on “**VoIP Call Out Routing”** table. The format is IP/phone number or URL/phone number. I.e. all the phone number can find a matched prefix rule will be forward to the IP, and phone number define on here.
Name	Specify a profile name. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.
Always	Always redirect forward to this IP (or URL)/phone number, original line will never ring and all incoming call will be forward to IP assigned here.
On Busy	Redirect forward to this IP (or URL)/phone number when busy, an incoming VoIP call will forward to IP assigned here when this line is busy.
No Answer	Redirect forward to this IP (or URL)/phone number when no answer over the time “**No Answer Sec**” , an incoming VoIP call will forward to IP assigned here when ring time over the defined on “**No Answer Sec**”.
No Answer Sec.	Defined the maximum wait seconds for redirect forward to another IP (or URL).
Delete	Delete this rule item on routing table.



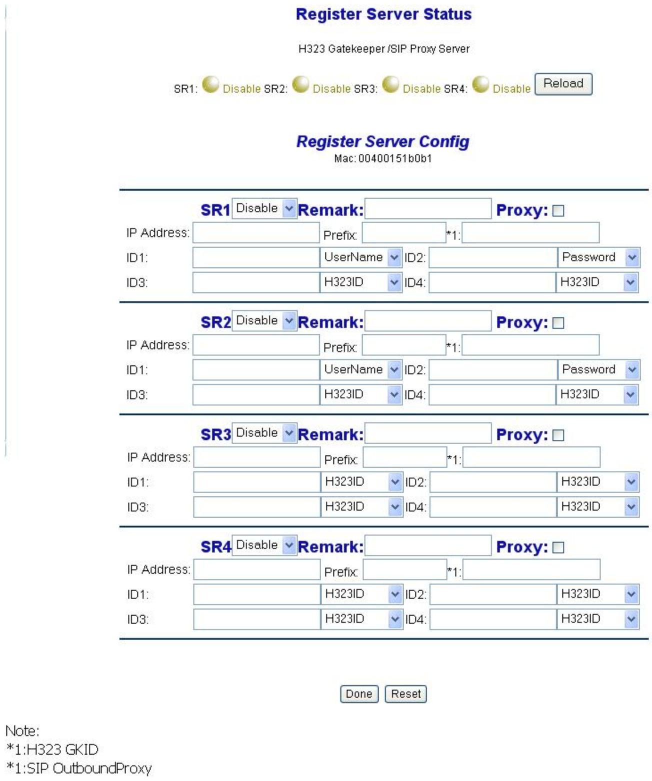
To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the modify button to take it effect. For store back to flash memory, please press “**Save Modification”** (Plaase refer to the “**Syetem Maintenance → Save Modification**”).

Register Server

If this VoIP router want to use GK/SIP proxy service to transfer the VoIP call, you can input the GK /SIP information here. The VoIP router can register to up to four GK/SIP proxy simultaneously.

Parameter Description	
Register Server Status	**Success:** Register successful. **Failure:** Register failure. **Disable:** disable register this GK/SIP proxy server
MAC	Display the MAC address of WAN on this VoIP router
SR1~SR4	**Enable:** Enable the VoIP router to register Server #1. **Disable:** Disable the VoIP router to register Server #1.
Remark	For Notify remark for this GK/SIP proxy server. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.
Proxy	Click to enable using GK/SIP proxy function. When enable, VoIP call will go through the GK/SIP proxy service. Please click here if your VoIP router is installed behind NAT or firewall without real IP. If you want use this function, please make sure your GK/SIP proxy has support the proxy function.
IP address:	Define the GK/SIP proxy server IP, user can input below format **IP address/URL for Gatekeepter server:** such as: **h323:172.16.0.100/ h323:h323.testcall.com.tw** **IP address/URL for SIP proxy server :** such as: **sip:172.16.0.200/sip:sip.testcall.com.tw**
Prefix	Specific the prefix number of this VoIP router service for register to GK/SIP proxy server.
ID1~ID4	Specific the ID of this VoIP router for register to GK/SIP proxy server **H.323:** register above ID as H.323 ID. **E.164:** register above ID as E.164 ID. **User Name:** register above ID as user name for GK/SIP proxy server. **Password:** register above ID as password for GK/SIP proxy server. There are four fields for user define the ID parameters, please reference the ID1 setting above.
***1:SIP OutboundProxy**	To make a call using SIP protocol with proxy server, input the server IP or domain name in the *1:SIP OutboundProxy field.



ⓘ Hint

When voice communication is established via **H.323 protocol**, please add a "h323:" in front of the IP address.
Such as: the GK IP address is 192.168.0.100, then input "h323:192.168.0.100" in the IP address.

When voice communication via the **SIP protocol**, please add a "sip:" in front of the IP address/URL.
Such as: the SIP-50 IP address is 192.168.0.50, then input "sip:192.168.0.50" in the IP address.

Please remember to press the "**Done"** button to take it effect. For store back to flash memory, please press "**Save Modification**" (Plaase refer to the "**Syetem Maintenance → Save Modification**").

WebCall

There is a embedded Web Call function within the VoIP router, The Web Call function let you call to the phone lines of this VoIP router with Web browser IE (Internet Explorer from Microsoft) . When a client PC uses browser open the embedded web this VoIP router, the embedded VoIP router will send the page with the parameters defined on "**VoIP Config → Web Call Setting**", and will launch the Net meeting within client PC windows OS. This function let a user PC with Internet connection to make a VoIP call to the lines connected to VoIP router. When user uses a browser to connect to the VoIP router, it will show the welcome Page:

Parameter Description	
Gateway IP	Show the IP of this VoIP router
Name	Select the name you want to make connect, this is defined on **Web Call** page. (Please refer to the "**VoIP Config → Web Call Setting → Web call**").
Call	Press to make a call.
Stop	Stop the call.

WebCall Config

This page let you define the welcome message, LOGO, call number when using Web Call function.

Web Call accept List:

Define the display name on select option during Web call.

Parameter Description	
Name	Name of selectable item during web call.
Number	Number of this selected item call out, when user select the name of this item rule, the number here will be used as the number for VoIP call In, and will check with the area code define on "**VoIP Config → Call Routing → VoIP Call In**", that mean you should have a matched item defined on "**VoIP Config → Call Routing → VoIP Call Out**".
Delete	Delete this rule item on routing table.
Stop	Stop the call.

To add new name item on Web Call accept List, please assign the number you want to insert before, input list item NAME then press ADD button to add it on the list. Then modify the necessary information on the r Web Call accept List.

Please remember to press the "**Modify"** button to take it effect. For store back to flash memory, please press "**Save Modification**"   (Plaase refer to the "**Syetem Maintenance → Save Modification**").

BWelcome page and banner Upload:

Define the welcome message and Logo for Web Call function:

Parameter Description	
User HTML Welcome Page	To upload a welcome message HTML file for display on Web Call function page, this page should be HTML file and there is a file size limitation, please press the "**Browse**" button to select the HTML file you want to upload and press "**Upload**" to Upload it.
User Welcome page banner	To upload a logo graphic file for display on Web Call function page, this graphic file should be name as "Welcome" only and there is no ext file

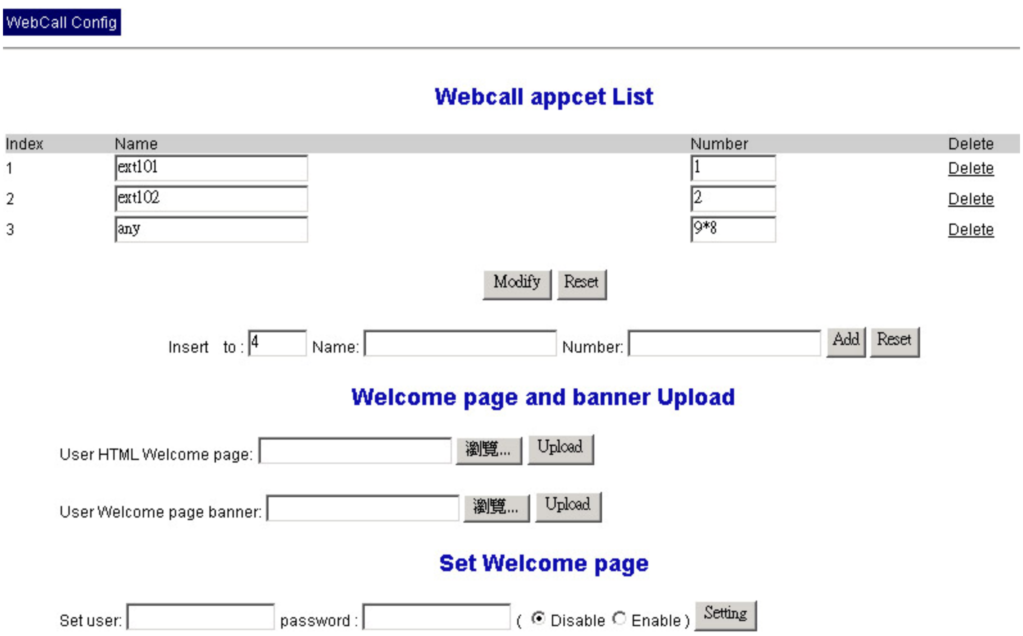
	name, please rename your logo graphic file(.bmp, .jpg, .gif) to "Welcome" before upload. There is a file size limitation. Please press the **Browse** button to select the "Welcome" file you want to upload and press **Upload** to Upload it.
Delete	Delete this rule item on routing table.
Stop	Stop the call.

Set Welcome page:

Set up the authorization check option for Web Call function. When Enable the authorization check, user need to input the valid user name and password to use the Web Call function.

- **Set User:** valid name for Web Call user
- **Password:** valid password for Web Call user.
- **Disable/Enable:** Disable or Enable username or password check for Web Call function.

When enable password check, user need to input the valid user name and password for Web Call.



Chapter 5

System Configurations

5

System Config

Bridge Mode Setting

This page allows you to disable/enable this device become bridge device or not. When it becomes a bridge device, bridge interface use LAN's IP address, LAN's subnet mask.

When working on Bridge Mode, the VoIP router will use only the LAN setting IP, The VoIP router will use the same LAN IP setting as WAN IP. That mean, When Bridge mode enable, the WAN connection setting will be ignored.

Date & Time

This page allows you to adjust the date & time settings in this router. The time settings are in 24-hour format. The router also uses the date and time to time stamp to log events.

Note: When you reset the router, you MUST adjust the date and time again.

Password

This page allows you to change the administration password used to manage this router for security reasons. o set this password, enter your current password in the Old Password field and then enter a New password in the New Password and Confirm New Password fields.

Note

The Default User name is "**admin**" and the password is "**123**" from factory. Press RESET button on rear panel over 5 seconds will cause the VoIP router reset to this default user name and password.

Basic Setup

This router comes with the built-in firewall based on the advanced technology of Stateful Packet Inspection to protect your network from being attacked by hackers. You can set up network access rules to decide if the network traffic is allowed to pass through (LAN-to-WAN and WAN-to-LAN) the firewall built inside the router.

In the following sections, you are able to configure firewall settings in this router. Some advanced knowledge or experiences in TCP/IP internet work are required.

Basic Settings: You can configure basic firewall settings in this router.

LAN-to-WAN Access Rules: You can define LAN-to-WAN network access rules which evaluate the network traffic's source IP address, destination IP address, and communication port to decide if it's allowed to pass through the firewall.

WAN-to-LAN Access Rules: You can define WAN-to-LAN network access rules which evaluate the network traffic's source IP address, destination IP address, and communication port to decide if it's allowed to pass through the firewall.

LAN to WAN Access Rules

This pages allows you to define LAN-to-WAN network access rules which evaluate the network traffic's source IP address, destination IP address, and communication port to decide if it's allowed to pass through the firewall.

By default, the stateful packet inspection module of this router allows all communications to the Internet that originates from the LAN. The behavior is defined by the default stateful packet inspection enabled in the router:

- Forward all sessions originating from the LAN to the Internet.
- Discard all sessions originating from the Internet to the LAN (Pleaes refer to the **"WAN-to-LAN Access Rules" at System Setup→ Firewall → WAN-to-LAN Access Rules**).

Additional access rules may be defined to extend or overwrite the default rules.

Note

- The ability to define network access rules is a very powerful management tool. Using a custom rule, it's possible to disable all firewall protection, creating holes in the firewall, or block all access to the Internet. Use with extreme caution when creating or deleting network access rules.
- Network access rules will not disable protection from Denial of Service (DoS) attacks, such as SYN Flood, Ping of Death, Port Scan, etc. However, it's possible to create vulnerabilities to attacks that exploit vulnerabilities in applications.

WAN to LAN Access Rules

This pages allows you to define WAN-to-LAN network access rules which evaluate the network traffic's source IP address, destination IP address, and communication port to decide if it's allowed to pass through the firewall.

By default, the stateful packet inspection module of this router blocks all traffic to the LAN that originates from the Internet. The behavior is defined by the default stateful packet inspection enabled in the router:

- Forward all sessions originating from the LAN to the Internet (Pleaes refer to the **"LAN-to-WAN Access Rules" at System Setup → Firewall → LAN-to-WAN Access Rules**).
- Discard all sessions originating from the Internet to the LAN.

Additional access rules may be defined to extend or overwrite the default rules.

Note

- The ability to define network access rules is a very powerful management tool. Using a custom rule, it's possible to disable all firewall protection, creating holes in the firewall, or block all access to the Internet. Use with extreme caution when creating or deleting network access rules.
- Network access rules will not disable protection from Denial of Service (DoS) attacks, such as SYN Flood, Ping of Death, Port Scan, etc. However, it's possible to create vulnerabilities to attacks that exploit vulnerabilities in applications.

Machine Status

This page display the Current Status of the VoIP router.

Dynamic DNS Setting

This section allows you to set up advanced features in this router. During the design stage, we have given much thought to making this router as convenient and easy to use as possible. However, some more advanced knowledge about TCP/IP might still be required.

Dynamic DNS: Each time the WAN address is changed, DDNS service will automatically update it to dyndns.org. You can register your account at :http://www.dyndns.org

DHCP Server Setting

This page allows you to set up configurations of DHCP server built in the router. The DHCP server of this router provides IP addresses, the subnet mask, the gateway address, and DNS server addresses to the LAN computers and devices dynamically. The default IP address space of this DHCP server is 192.168.0.x, with subnet mask 255.255.255.0, and the default gateway of this network is the IP address of this router (192.168.0.1).

It's highly recommended you use this router as the DHCP server; unless you already have a DHCP server on the network.

The DHCP server comes with two default IP lease ranges. To add a new dynamic IP range for lease, click the "**Show Current IP Ranges**" section.

To view the current dynamic IP assignments from the DHCP server, click "**Show IP Lease Table (Show DHCP leases)"**.

To assign a fixed-IP for a certain host on private network, click "**Show Fixed-IP Table**".

Note

When any change is made on this page, you MUST restart all PCs to update their TCP/IP settings from this DHCP server.

Static Routing

This page mainly allows you to define a static routing entry in the internal routing table of the router. If the private LAN has internal routers, their addresses and network information will need to be entered into this router to find the correct data path when it routes network packets. Static routes are generally used if the LAN are segmented into subnets, either for size or practical considerations.

Most of users who are using the whole IP address space without sub networks don't have to enter any entry in this table. The router automatically updates its internal routing table and dynamically notifies other routers on the network by sending out RIP (Routing Information Protocol) information. This router supports RIP I and RIP II standards.

To add a new static routing path, click "**View or Add Static Routing Table**" link.

Note

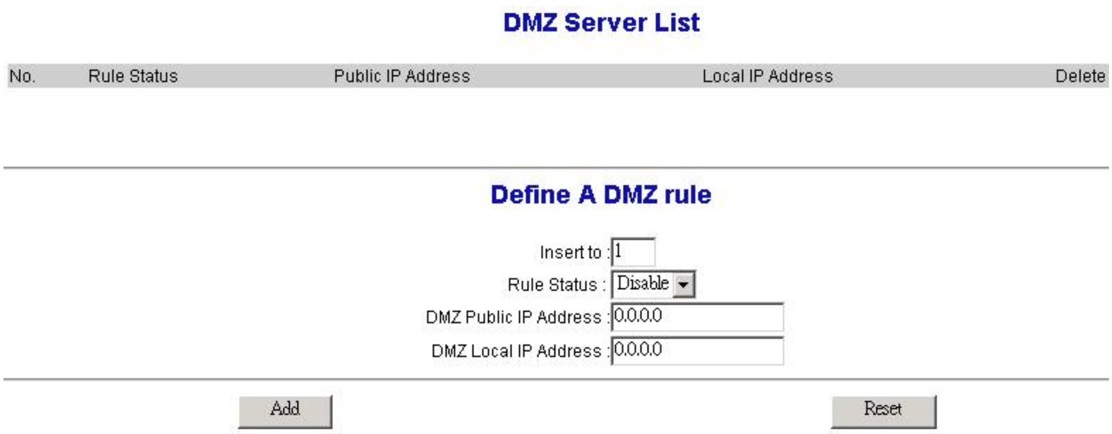
Adding incorrect routing information can affect the connection, a local host, or the whole private network. You must have experience working with routing tables before using this option.

Virtual Server

This page allows you to map a TCP or a UDP port of the router to a host which actually deals with requests on the private network.

DMZ

This page let you set up the DMZ service on the VoIP router.



System Maintenance

This page let you backup / Restore all of your configuration parameters on the VoIP router. It is very good idea to back up all of your VoIP router configuration parameters after install.

Configurations

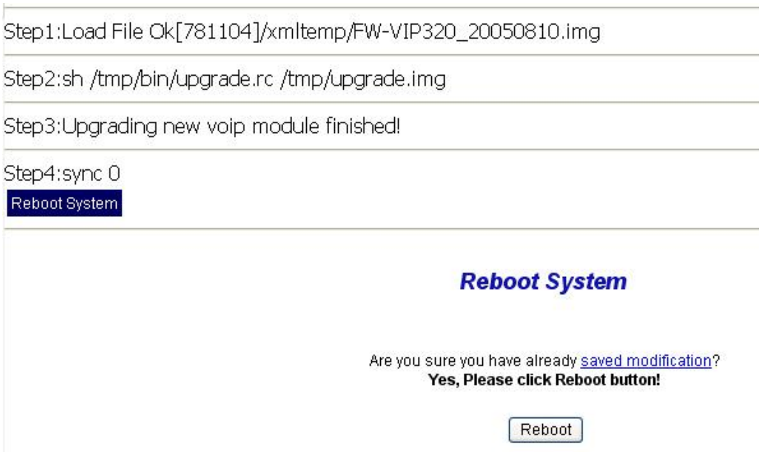
To Backup, press Download setting backup file, and input the file name you want and file location to save.

To Restore, press the Browse button the select the backup configuration parameters file to upload then press Restore . After you upload the file, Press “**Saved modification”** to save your current configuration to Flash ROM (Usually used to save currently WAN configuration).After save, please remember to “**Reboot**” the VoIP router to let the restored parameters take effective.

Firmware Upgarade Procerdure:

Pleaes download the latest firmware to a PC firset, and browse to the **“Backup/Restore --> Configurations”** menu, and click on the **”Browse”** icon to select the file, once the firmware file is entered, please click on the **”Restore”** icon to proceed with the updating process.

After process completed, please click on the **”Reboot”** button below:



ⓘ Hint

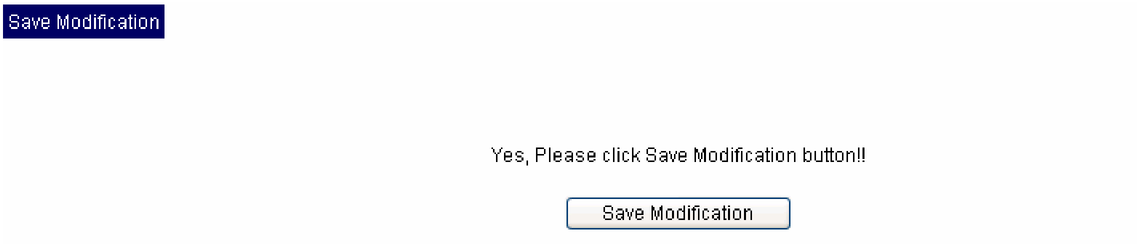
Never power off the VoIP router when restoring machine configuration file or upgrading the firmware, the machine will be damaged permanently.

Reboot System

Use the Reboot button on this page to reboot your VoIP router, before you reboot, please make sure you have to press the “**Saved modification**” to save your current configuration to Flash ROM, otherwise all the change will be disappear after reboot.

Save Modification to Flash Memory

Most of the VoIP router parameters will take effective after you modify, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the VoIP router, to save the parameters into Flash ROM and let it take effective forever, please remember to press the Save Modification button after you modify the parameters.



Appendix A Voice communications

There are several ways to make calls to desired destination in VIP-280. In this chapter, we'll lead you step by step to establish your first voice communication via web browsers operations.

Default Configuration

Without any configuration, your VIP-280 is come with following basic information.

Line Setting:

Default line number: 201, 202

Setup Help | Line Setting | Tone Setting

Port	Interface Name		Line Number	TxGain	RxGain	InBound	OutBound	HotLine
1	FXS	1	201	0 db	0 db	Enable	Enable	Disable
2	FXS	2	202	0 db	0 db	Enable	Enable	Disable

Modify Reset

VoIP Call Out:

Default line number: 201, 202

Setup Help | VoIP Call Out | VoIP Call In | Call Setup | Call Forwarding | Authorisation

MaxDigits: 20 FirstDigitTime(Sec): 30 OtherDigitTime(Sec): 5

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	Line1	201			sip:172.16.0.1				Delete
2	Line2	202			sip:172.16.0.1				Delete

VoIP Call In:

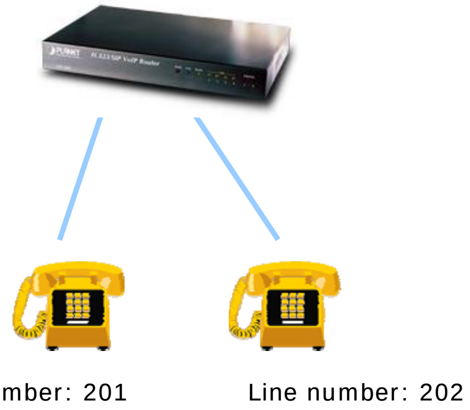
Default line number: 201, 202

Setup Help | VoIP Call Out | VoIP Call In | Call Setup | Call Forwarding

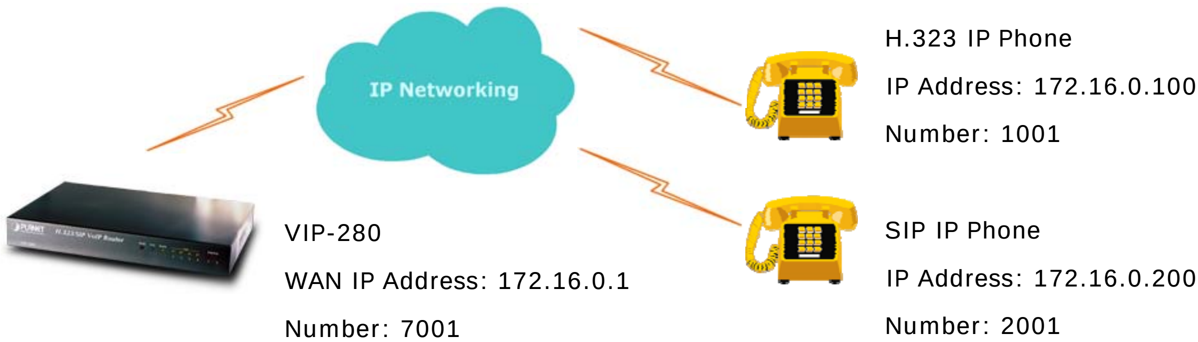
Index	Area Code	Strip	Prefix	Maximum	Minimum	From	To	LineNo	Server	-Ans.	Alert	Profile	Forward	Delete
1	201					1	1	☐		Before	1			Delete
2	202					2	2	☐		Before	0			Delete

Supposing you have one VIP-280 connects to four telephones, just pick up phone 1 and dial ‘202’, phone 2 should rings.

☎ ☝202 ⇨ VIP-280 ⇨ local port #2⇨ ☎)))



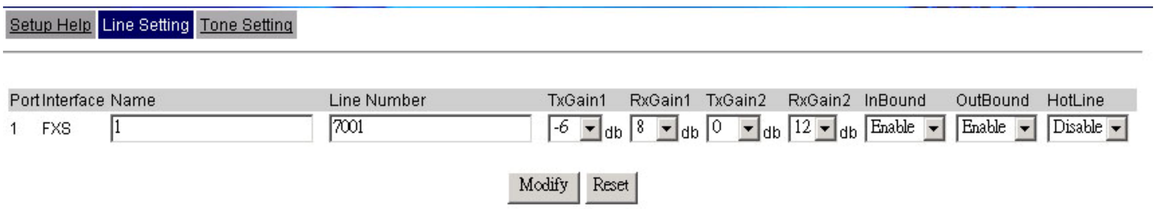
Peer-to-Peer (P2P) mode



VIP-280 configurations:

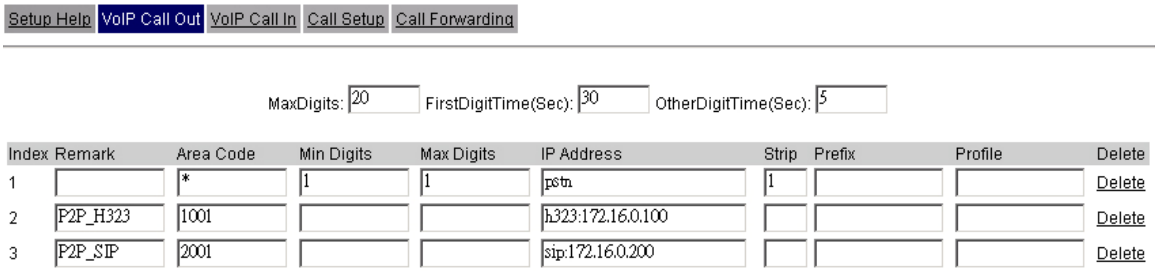
STEP 1:

Please log in machine via web browser, and select **Line Setting** in the **Line config** menu. In this Line Setting page, please insert the telephone number assigned to this line**,** and then the sample configuration screen is shown below (in this sample, we’re using number **7001** for incoming calls).



STEP 2:

Select **VoIP Call Out** in the **Call Routing** menu; insert the values of the index number, Area Code and IP Address on the VoIP call out routing table for outgoing calls. The sample configuration screen is shown below.



Setup Help | VoIP Call Out | VoIP Call In | Call Setup | Call Forwarding

MaxDigits: 20 FirstDigitTime(Sec): 30 OtherDigitTime(Sec): 5

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1		*	1	1	pstn	1			Delete
2	P2P_H323	1001			h323:172.16.0.100				Delete
3	P2P_SIP	2001			sip:172.16.0.200				Delete

ⓘ Hint

```
When the calling party is an H.323 device, please add
a "h323:" in front of the IP address.
Such as: the destination H.323 device is 172.16.0.100, then
input "h323:172.16.0.100" in the IP address column of
VIP-320/VIP-280 VoIP Callout setting page

When the calling party is a SIP device, please add a "sip:"
in front of the IP address.
Such as: the destination SIP device is 172.16.0.200, then
input "sip:172.16.0.200" in the IP address field.
```

STEP 3:

After the settings for the remote calling party, you may dial number 1001 to connect to the H.323 IP phone, and number 2001 to connect to the SIP IP phone.

ⓘ Hint

```
If you're using the VIP-280, you may dial or receive the
H.323 and the SIP calls at the same time.
```

Voice communication via SIP proxy server –SIP50

- VIP-280 IP Address: 172.16.0.28
  Line Number: 280
- VIP-320 IP Address: 172.16.0.32
  Line Number: 320

Machine configurations on the VIP-280:

STEP 1:

Please log in machine via web browser, and select **Register Server** setting in the **VoIP Config** menu. In this setting page, please insert the account/password information, and then the sample configuration screen is shown below (in this sample, we're using the SIP-50 as the registration server).

ID1: **280** is the phone number that will register to the SIP-50

ID2: **vip280** is the name registered in SIP-50

ID3: the password for account vip280

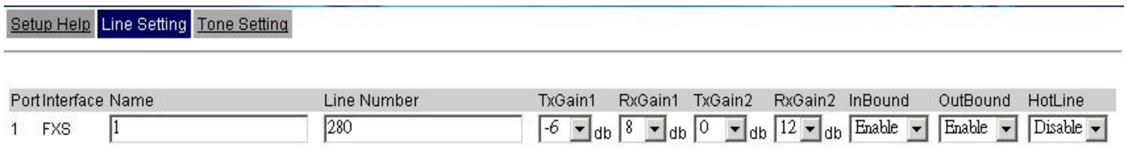
ⓘ Hint

When voice communication is established via "**Gatekeeper**", please add a "**h323:**" in front of the IP address.
Such as: the GK IP address is 192.168.0.100, then input "**h323:192.168.0.100**" in the IP address.

When voice communication via the SIP proxy server, please add a "**sip:**" in front of the IP address/URL.
Such as: the SIP-50 IP address is 192.168.0.50, then input "**sip:192.168.0.50**" in the IP address.

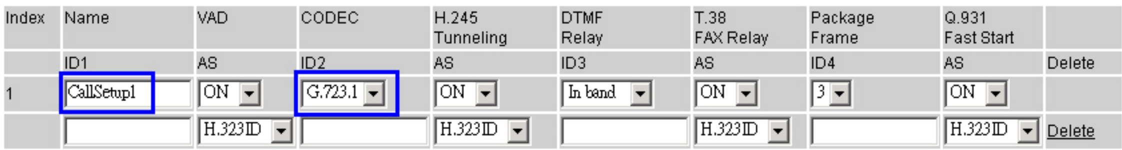
STEP 2:

Select **Line Setting** in the **Line config** menu. In this Line Setting page, please insert the telephone number assigned to this line**,** and then the sample configuration screen is shown below (in this sample, we're using number **280** for incoming calls).



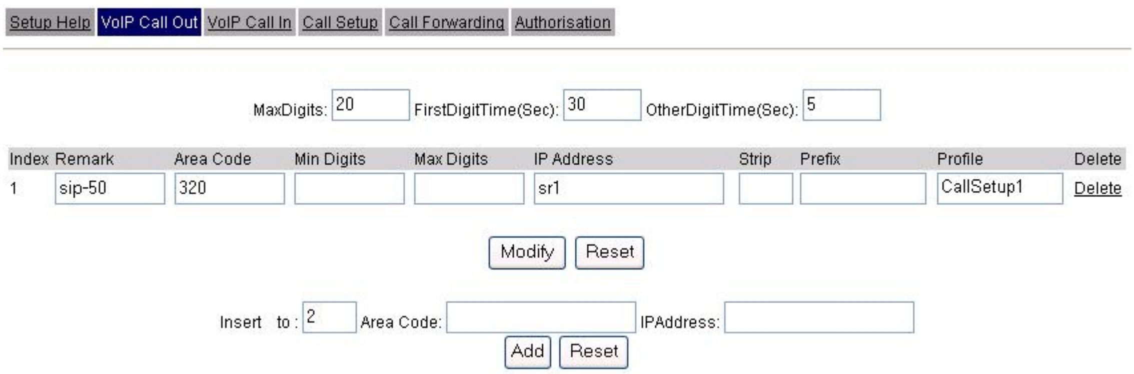
STEP 3:

If wants to assign the individual voice codec (G.723.1/G.729/G.711) to establish the voice communications, please browse the **Call Setup** in the **Call Routing** menu, and refer to the following configuration illustrate:



STEP 4:

Select VoIP Call Out in the Call Routing menu; insert the values of the index number, Area Code, IP Address and Profile on the VoIP call out routing table for outgoing calls. The sample configuration screen is shown below. The IP address "sr1" is the alias name of register server 1. Area Code "320" is the number for VIP-320.



IP Address: **sr1** is the alias name of register server 1.

Profile: **CallSetup1** is the profile that created in STEP 3.

Area Code: **320** is the number to call the VIP-320. Of course, if there are more phones registered to SIP-50, say, there are number 321, 322, 323… You can just use "**3**" as the area code, whenever you dial **3**xx, such as 322, 323, and then it will check SIP-50 for the available phones.

Remark: **sip-50** is the explaination of this index.

Note: please remember to press the **Save Modification** button after you modify the parameters.

STEP 5:

Repeat the same configuration steps on the VIP-320, and check the machine registration status, make sure the registrations are completed.

===

Test the scenario:

To verify the VoIP communication, you may make calls from SIP client (VIP-280) 280 to the SIP client (VIP-320) 320 or reversely make calls from SIP client (VIP-320) 320 to the SIP client (VIP-280) 280.

==

Appendix B VIP-280 Specifications

Product	H.323/ SIP VoIP Router
Model	VIP-280
Hardware	
WAN	1 x 10/100Mbps RJ-45 port
LAN	4 x 10/100Mbps RJ-45 port
FXS	2 x RJ-11 connection
Standards and protocol	
Standard	H.323 version v2/v3,H.323 Fast start, and H.245 DTMF relay, SIP 2.0 (RFC3261)
Voice codec	G.723.1 (6.3k/5.3k), G.729A, G.711 (A-law/U-law)
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Dynamic Jitter Buffer
Supplementary services	Call transferring between DECT handsets
Protocols	SIP 2.0 (RFC-3261), H.323, TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE
Internet features	Built in NAT firewall, DoS (Denial of Service) protection SPI (Stateful Packet Inspection) firewall Policy-based LAN/WAN access control Virtual server, DMZ, Remote administrator authentication
Network and Configuration	
Access Mode	Static IP, PPPoE, DHCP
Management	Web
Dimension (W x D x H)	222 x 146 x 29 mm
Operating Environment	0~40 degree C, 10~95% humidity
Power Requirement	12V DC
EMC/EMI	CE, FCC Class B